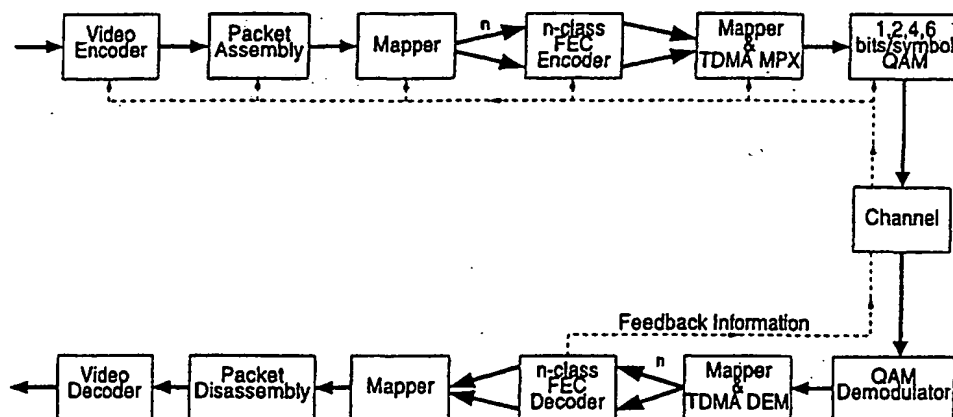




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(54) Title: BURST-BY-BURST ADAPTIVE SINGLE-CARRIER DATA TRANSMISSION



(57) Abstract

The performance benefits of burst-by-burst adaptive modulation are studied, employing a higher-order modulation scheme, when the channel is favourable, in order to increase the system's bits per symbol capacity and conversely, invoking a more robust, lower order modulation scheme, when the channel exhibits inferior channel quality. It is shown that due to the described adaptive modem mode switching regime a seamless multimedia source-signal representation quality – such as video or audio quality – versus channel quality relationship can be established, resulting in a near-unimpaired multimedia source-signal quality right across the operating channel Signal-to-Noise Ratio (SNR) range. The main advantage of the described technique is that irrespective of the prevailing channel conditions, the transceiver achieves always the best possible source-signal representation quality – such as video or audio quality – by automatically adjusting the achievable bitrate and the associated multimedia source-signal representation quality in order to match the channel quality experienced. This is achieved on a near-instantaneous or burst-by-burst adaptive basis under given propagation conditions in order to cater for the effects of path-loss, fast-fading, slow-fading, dispersion, co-channel interference, etc. Furthermore, when the mobile is roaming in a hostile out-doors – or even hilly terrain – propagation environment, typically low-order, low-rate modem modes are invoked, while in benign indoor environments predominantly the high-rate, high source-signal representation quality modes are employed.

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Title of the Invention

Burst-by-burst Adaptive Single-carrier Data Transmission

1 Background of the Invention

The invention relates to data transmission, more specifically to transmission in packets or bursts.

In contrast to the *burst-by-burst reconfigurable wideband* multimedia transceivers described in this document, the term *statically reconfigurable* found in this context in the literature refers to multimedia transceivers that cannot be near-instantaneously reconfigured. More explicitly, the previously proposed *statically reconfigurable* video transceivers were reconfigured on a long-term basis under the base station's control, invoking for example in the central cell region - where benign channel conditions prevail - a less robust, but high-throughput modulation mode, such as 4 bit/symbol Quadrature Amplitude Modulation (16QAM), which was capable of transmitting a quadruple number of bits and hence ensured a better video quality. By contrast, a robust, but low-throughput modulation mode, such as 1 bit/symbol Binary Phase Shift Keying (BPSK) can be employed near the edge of the propagation cell, where hostile propagation conditions prevail. This prevented a premature hand-over at the cost of a reduced video quality.

The philosophy of the fixed, but programable-rate proprietary video codecs and statically reconfigurable multi-mode video transceivers presented by Streit et al in References [1]-[4] was that irrespective of the video motion activity experienced, the specially designed video codecs generated a constant number of bits per video frame. For example, for videophony over the second-generation Global System of Mobile Communications known as the GSM system at 13 kbps and assuming a video scanning rate of 10 frames/s, 1300 bits per video frame have to be generated. Specifically, two families of video codecs were designed, one refraining from using error-sensitive run-length coding techniques and exhibiting the highest possible error resilience and another, aiming for the highest possible compression ratio. This fixed-rate approach had the advantage of requiring no adaptive feedback controlled bitrate fluctuation smoothing buffering and hence exhibited no objectionable video latency or delay. Furthermore, these video codecs were amenable to video telephony over fixed-rate second-generation mobile radio systems, such as the GSM.

The fixed bitrate of the above proprietary video codecs is in contrast to existing standard video codecs, such as the Motion Pictures Expert Group codecs known as MPEG1 and MPEG2 or the ITU's H.263 codec, where the time-variant video motion activity and the variable-length coding techniques employed

31 result in a time-variant bitrate fluctuation and a near-constant perceptual video quality. This time-variant
32 bitrate fluctuation can be mitigated by employing adaptive feed-back controlled buffering, which poten-
33 tially increases the latency or delay of the codec and hence it is often objectionable for example in inter-
34 active videophony. The schemes presented by Streit et al in Références [1]-[4] result in slightly variable
35 video quality at a constant bitrate, while refraining from employing buffering, which again, would result
36 in latency in interactive videophony. A range of techniques, which can be invoked, in order to render the
37 family of variable-length coded, highly bandwidth-efficient, but potentially error-sensitive class of stan-
38 dard video codecs, such as the H.263 arrangement, amenable to error-resilient, low-latency interactive
39 wireless multimode videophony was summarised in [5]. The adaptive video rate control and packetisa-
40 tion algorithm of [5] generates the required number of bits for the burst-by-burst adaptive transceiver,
41 depending the on the capacity of the current packet, as determined by the current modem mode. Fur-
42 ther error-resilient H.263-based schemes were contrived for example by Färber, Steinbach and Girod
43 at Erlangen University [6], while Sadka, Eryurtlu and Kondo [7] from Surrey University proposed a
44 range of improvements to the H.263 scheme. Following the above portrayal of the prior art in both video
45 compression and statically reconfigurable narrowband modulation, let us now consider the philosophy of
46 wideband burst-by-burst adaptive quadrature amplitude modulation (AQAM) in more depth.

47 In burst-by-burst adaptive modulation a higher-order modulation scheme is invoked, when the channel
48 is favourable, in order to increase the system's bits per symbol capacity and conversely, a more robust
49 lower order modulation scheme is employed, when the channel exhibits inferior channel quality, in order
50 to improve the mean Bit Error Ratio (BER) performance. A practical scenario, where adaptive modula-
51 tion can be applied is, when a reliable, low-delay feedback path is created between the transmitter and
52 receiver, for example by superimposing the estimated channel quality perceived by the receiver on the
53 reverse-direction messages of a duplex interactive channel. The transmitter then adjusts its modem mode
54 according to this perceived channel quality.

55 Recent developments in adaptive modulation over a narrow-band channel environment have been pi-
56 oneered by Webb and Steele [9], where the modulation adaptation was utilized in a Digital European
57 Cordless Telephone - like (DECT) system. The concept of variable rate adaptive modulation was also
58 advanced by Sampei *et al* [12, 17], showing promising advantages, when compared to fixed modula-
59 tion in terms of spectral efficiency, BER performance and robustness against channel delay spread. In
60 another paper, the numerical upper bound performance of adaptive modulation in a slow Rayleigh flat-
61 fading channel was evaluated by Torrance *et al* [10] and subsequently, the optimization of the switching
62 threshold levels using Powell minimization was used in order to achieve a targeted performance [11, 18].
63 In addition, adaptive modulation was also studied in conjunction with channel coding and power control

64 techniques by Matsuoka *et al* [12] as well as Goldsmith *et al*. [13]-[15].
65 In the narrow-band channel environment, the quality of the channel was determined by the short term
66 Signal to Noise Ratio (SNR) of the received burst, which was then used as a criterion in order to choose
67 the appropriate modulation mode for the transmitter, based on a list of switching threshold levels. *l*, [9,
68 10]. However, in a wideband environment, this criterion is not an accurate measure for judging the quality
69 of the channel, where the existence of multi-path components produces not only power attenuation of the
70 transmission burst, but also intersymbol interference. Subsequently, a new criterion has to be defined to
71 estimate the wideband channel quality in order to choose the appropriate modulation scheme.

72 2 Summary of the Invention

73 Particular and preferred aspects of the invention are set out in the accompanying independent and depen-
74 dent claims. Features of the dependent claims may be combined with those of the independent claims as
75 appropriate and in combinations other than those explicitly set out in the claims.
76 The performance benefits of burst-by-burst adaptive modulation are described, employing a higher-order
77 modulation scheme, when the channel is favourable, in order to increase the system's bits per symbol
78 capacity and conversely, invoking a more robust, lower order modulation scheme, when the channel
79 exhibits inferior channel quality. It is shown that due to the described adaptive modem mode switch-
80 ing regime a seamless multimedia source-signal representation quality - such as video or audio quality -
81 versus channel quality relationship can be established, resulting in a near-unimpaired multimedia source-
82 signal quality right across the operating channel Signal-to-Noise Ratio (SNR) range. The main advan-
83 tage of the described technique is that irrespective of the prevailing channel conditions, the transceiver
84 achieves always the best possible source-signal representation quality - such as video or audio quality - by
85 automatically adjusting the achievable bitrate and the associated multimedia source-signal representation
86 quality in order to match the channel quality experienced. This can be achieved on a near-instantaneous or
87 burst-by-burst adaptive basis under given propagation conditions in order to cater for the effects of path-
88 loss, fast-fading, slow-fading, dispersion, co-channel interference, etc. Furthermore, when a mobile is
89 roaming in a hostile out-doors - or even hilly terrain - propagation environment, typically low-order,
90 low-rate modem modes are invoked, while in benign indoor environments predominantly the high-rate,
91 high source-signal representation quality modes are employed.

3 Brief Description of the Drawings

For a better understanding of the invention and to show how the same may be carried into effect reference is now made by way of example to the accompanying drawings, in which:

List of Figures

- 1 Signalling scenarios in adaptive modems between a Mobile Station (MS) and a Base Station (BS)
- 2 Reconfigurable transceiver schematic
- 3 Normalized channel impulse response for the COST 207 four-path Typical Urban channel.
- 4 Transmission burst structure of the FMA1 non-spread data burst mode of the FRAMES proposal
- 5 Adaptive burst-by-burst modem in operation for an average channel SNR of 20dB, where the modulation mode switching is based upon the SNR estimate at the output of the equaliser, using the channel parameters defined in Table 1.
- 6 PDF of the adaptive modem being in a particular modulation mode versus channel SNR.
- 7 Transmission FER (or packet loss ratio) versus Channel SNR comparison of the four fixed modulation modes (BPSK, 4QAM, 16QAM, 64QAM) and that of the adaptive burst-by-burst modem (AQAM). AQAM is shown with a realistic one TDMA frame delay between channel estimation and mode switching, and also with a zero delay version for indicating the upper bound performance. The channel parameters were defined in Table 1.
- 8 Transmission FER (or packet loss ratio) versus Channel SNR comparison of the four fixed modulation modes (BPSK, 4QAM, 16QAM, 64QAM) with 5% FER switching and adaptive burst-by-burst modem (AQAM). AQAM is shown with a realistic one TDMA frame delay between channel estimation and mode switching, and a zero delay version is included as an upper bound. The channel parameters were defined in Table 1.
- 9 Video bitrate versus channel SNR comparison of the four fixed modulation modes (BPSK, 4QAM, 16QAM, 64QAM) and adaptive burst-by-burst modem (AQAM). AQAM is shown with a realistic one TDMA frame delay between channel estimation and mode switching, and also as a zero delay version for indicating the upper bound. The channel parameters were defined in Table 1.

- 122 10 Decoded video quality (PSNR) versus channel SNR comparison of the four fixed mod-
123 ulation modes (BPSK, 4QAM, 16QAM, 64QAM) with 5% transmission FER switching
124 and that of the adaptive burst-by-burst modem (AQAM). AQAM is shown with a realis-
125 tic one TDMA frame delay between channel estimation and mode switching, and a zero
126 delay version for indicating the upper bound. The channel parameters were defined in
127 Table 1.
- 128 11 Decoded video quality (PSNR) versus channel SNR for the adaptive burst-by-burst mo-
129 dem (AQAM). AQAM is shown with a realistic one TDMA frame delay between channel
130 estimation and mode switching, and a zero delay version indicating the upper bound. Re-
131 sults are shown for three video sequences using the channel parameters that were defined
132 in Table 1.
- 133 12 Decoded video quality (PSNR) versus transmission FER (or packet loss ratio) compar-
134 ison of the four fixed modulation modes (BPSK, 4QAM, 16QAM, 64QAM) and that
135 of the adaptive burst-by-burst modem (AQAM). AQAM is shown with a realistic one
136 TDMA frame delay between channel estimation and mode switching, and a zero delay
137 version indicating the upper bound. The channel parameters were defined in Table 1. . . .
- 138 13 Transmission FER (or packet loss ratio) versus Channel SNR comparison of the fixed
139 BPSK modulation mode and the adaptive burst-by-burst modem (AQAM) for the three
140 sets of switching thresholds described in Table 4. AQAM is shown with a realistic one
141 TDMA frame delay between channel estimation and mode switching. The channel pa-
142 rameters were defined in Table 1.
- 143 14 Video bitrate versus channel SNR comparison for the adaptive burst-by-burst modem
144 (AQAM) with a realistic one TDMA frame delay between channel estimation and mode
145 switching for the three sets of switching thresholds as described in Table 4. The channel
146 parameters were defined in Table 1.

147 4 Detailed Description

148 4.1 General Introduction to Adaptive Modem Mode Signalling Scenarios

149 AQAM transmission parameter adaptation is an action of the transmitter in response to time-varying
150 channel conditions. It is only suitable for duplex communication between two stations, since the trans-
151 mission parameter adaptation relies on some form of channel estimation and signalling. In order to
152 efficiently react to the changes in channel quality, the following steps have to be taken:

- 153 • *Channel quality estimation:* In order to appropriately select the transmission parameters to be
154 employed for the next transmission, a reliable prediction of the channel quality during the next
155 active transmit timeslot is necessary.
- 156 • *Choice of the appropriate parameters for the next transmission:* Based on the prediction of the
157 expected channel conditions during the next timeslot, the transmitter has to select the appropriate
158 modulation schemes for the subcarriers.
- 159 • *Signalling or blind detection of the employed parameters:* The receiver has to be informed, as
160 to which set of demodulator parameters to employ for the received packet. This information can
161 either be conveyed within the packet, at the cost of loss of useful data bandwidth, or the receiver
162 can attempt to estimate the parameters employed at the transmitter by means of blind detection
163 mechanisms.

164 Depending on the channel characteristics, these operations can be performed at either of the duplex
165 stations, as shown in Figure 1. If the channel is reciprocal, then the channel quality estimation for each
166 link can be extracted from the reverse link, and we refer to this regime as open-loop adaptation. In this
167 case, the transmitter needs to communicate the transmission parameter set to the receiver (Figure 1(a)),
168 or the receiver can attempt blind detection of the transmission parameters employed (Figure 1(c)).
169 If the channel is not reciprocal, then the channel quality estimation has to be performed at the receiver
170 of the link. In this case, the channel quality measure or the set of requested transmission parameters is
171 communicated to the transmitter in the reverse link (Figure 1(b)). This mode is referred to as closed-loop
172 adaptation.

173 4.2 A Specific Embodiment of a Video Transceiver

174 The schematic of the whole system is depicted in Figure 2. In the described system the wideband channel-
175 induced degradation is combated not only by the employment of adaptive modulation but also by equal-
176 ization, where the equalization process will eliminate most of the intersymbol interference based on a

177 Channel Impulse Response (CIR) estimate derived using the channel sounding midamble and conse-
178 quently, the signal to noise and residual interference ratio at the output of the equalizer is calculated.
179 We note, however that the above adaptive methodology can also be extended to employing burst-by-
180 burst adaptive channel coding associated with different-strength error correction codecs in the different
181 transceiver modes of operation.

182 4.3 Channel quality metrics

183 The most reliable channel quality estimate is the bit error rate (BER), since it reflects the channel quality,
184 irrespective of the source or the nature of the quality degradation.

185 Firstly, the BER can be estimated with a certain granularity or accuracy, provided that the system entails
186 a channel decoder or - synonymously - Forward Error Correction (FEC) decoder employing algebraic
187 decoding.

188 Secondly, if the system contains a soft-in-soft-out (SISO) channel decoder, the BER can be estimated
189 with the aid of the Logarithmic Likelihood Ratio (LLR), evaluated either at the input or the output of the
190 channel decoder. A particularly attractive way of invoking LLRs is employing powerful turbo codecs,
191 which provide a reliable indication of the confidence associated with a particular bit decision in the
192 context of LLRs. The LLR is defined as the ratio of the probabilities of a specific bit being binary zero
193 or one. Again, this measure can be evaluated at both the input and the output of the turbo channel codecs
194 and both of them can be used for channel quality estimation.

195 Thirdly, in the event that no channel encoder / decoder (codec) is used in the system, the channel quality
196 expressed in terms of the BER can be estimated with the aid of the mean-squared error (MSE) at the
197 output of the channel equaliser or the closely related metric, the Pseudo-Signal-to-noise-ratio (Pseudo-
198 SNR). The MSE or pseudo-SNR at the output of the channel equaliser have the important advantage
199 that they are capable of quantifying the severity of the inter-symbol-interference (ISI) and/or Co-channel
200 Interference experienced, in other words quantifying the Signal to Interference plus Noise Ratio (SINR).

201 4.3.1 Pseudo-SNR Embodiment

202 A specific embodiment based on the above-mentioned pseudo-SNR is now described in more depth.
203 Employing the pseudo-SNR has the advantage that it is generally applicable, regardless of whether or
204 not a channel codec is present.

We found that the residual channel-induced inter-symbol-interference (ISI) at the output of the decision
feedback equaliser (DFE) is near-Gaussian distributed and that if there are no decision feedback errors,

the pseudo-SNR at the output of the DFE, γ_{dfe} can be calculated as [8]:

$$\begin{aligned}\gamma_{dfe} &= \frac{\text{Wanted Signal Power}}{\text{Residual ISI Power} + \text{Effective Noise Power}} \\ &= \frac{E \left[|S_k \sum_{m=0}^{N_f-1} C_m h_m|^2 \right]}{\sum_{q=-(N_f-1)}^{-1} E \left[|f_q S_{k-q}|^2 \right] + N_o \sum_{m=0}^{N_f-1} |C_m|^2}\end{aligned}\quad (1)$$

where C_m and h_m denotes the DFE's feed-forward coefficients and the channel impulse response, respectively. The transmitted signal and the noise spectral density is represented by S_k and N_o . Lastly, the number of DFE feed-forward coefficients is denoted by N_f . By utilizing the pseudo-SNR at the output of the equalizer, we are ensuring that the system performance is optimised by employing equalization and adaptive quadrature amplitude modulation (AQAM) in a wideband environment according to the following switching regime:

$$\text{Modulation Mode} = \begin{cases} BPSK & \text{if } \gamma_{DFE} < f_1 \\ 4QAM & \text{if } f_1 < \gamma_{DFE} < f_2 \\ 16QAM & \text{if } f_2 < \gamma_{DFE} < f_3 \\ 64QAM & \text{if } \gamma_{DFE} > f_3, \end{cases} \quad (2)$$

205 where $f_n, n = 1 \dots 3$ are the pseudo-SNR thresholds levels, which are set according to the system's integrity
206 requirements.

207 In contrast to the narrowband, statically reconfigured multimode systems of [1]-[4] constituting the state-
208 of-the-art, the present embodiment invokes wideband, near-instantaneously reconfigured or burst-by-
209 burst adaptive channel-equalised modulation, in order to achieve the best possible multimedia source-
210 signal representation quality - for example video quality - when transmitting over arbitrarily time-variant
211 channels, exhibiting short-term and/or long-term channel quality variations. These variations can be
212 due to the effects of path-loss, fast-fading, slow-fading, dispersion, co-channel interference, etc. Further-
213 more, when the mobile is roaming in a hostile out-doors - or even hilly terrain - propagation environment,
214 typically low-order, low-rate modem modes are invoked, while in benign indoor environments predomi-
215 nantly the high-rate, high video quality modes are employed.

216 It is an important element of the system that when the binary BCH channel codes or FEC codes protect-
217 ing the video stream are overwhelmed by the plethora of transmission errors, the embodiment refrains
218 from decoding the video packet in order to prevent error propagation through the reconstructed frame
219 buffer [5]. Instead, these corrupted packets are dropped and the reconstructed frame buffer will not be
220 updated, until the next packet replenishing the specific video frame area arrives. The associated video

Parameter	Value
Carrier Frequency	1.9GHz
Vehicular Speed	30mph
Doppler frequency	85Hz
Normalised Doppler frequency	3.27×10^{-5}
Channel type	COST 207 Typical Urban (see Figure 3)
Number of paths in channel	4
Data modulation	Adaptive QAM (BPSK, 4-QAM, 16-QAM, 64-QAM)
Receiver type	Decision Feedback Equalizer Number of Forward Filter Taps = 35 Number of Backward Filter Taps = 7

Table 1: Modulation and channel parameters

performance degradation is fairly minor for packet dropping or frame error rates (FER) below about 5%. These packet dropping events are signalled to the remote decoder by superimposing a strongly protected one-bit packet acknowledgement flag on the reverse-direction packet, as outlined in [5]. In the embodiment we also invoked the adaptive rate control and packetisation algorithm of [5], supporting constant Baud-rate operation.

As a specific example of the burst-by-burst adaptive multimedia system we used 176x144 pixel so-called QCIF-resolution, 30 frames/s video sequences encoded at bitrates resulting in high perceptual video quality, in order to demonstrate the performance advantages of the video transceiver. Table 1 shows the modulation- and channel parameters employed, noting again that the associated principles are applicable in the context of a whole range of other system parameters. The COST207 four-path typical urban (TU) channel model was used in quantifying the associated system performance and its impulse response is portrayed in Figure 3. As an example, we used the Pan-European FRAMES proposal as the basis for our wideband transmission system, the frame structure of which is shown in Figure 4. Employing the FRAMES Mode A1 (FMA1) so-called non-spread data burst mode required a system bandwidth of 3.9MHz, when assuming a modulation excess bandwidth of 50%. A range of other system parameters are shown in Table 2.

The specific example of the video transceiver - which is used to demonstrate the advantages of the system concept - is based on the H.263 video codec. The video coded bitstream was protected by binary Bose-

Features	Value
Multiple access	TDMA
No. of Slots/Frame	16
TDMA frame length	4.615ms
TDMA slot length	288 μ s
Data Symbols/TDMA slot	684
User Data Symbol Rate (KBd)	148.2
System Data Symbol Rate (MBd)	2.37
Symbols/TDMA slot	750
User Symbol Rate (KBd)	162.5
System Symbol Rate (MBd)	2.6
System Bandwidth (MHz)	3.9
Eff. User Bandwidth (kHz)	244

Table 2: Generic system features of the reconfigurable multi-mode video transceiver, using the non-spread data burst mode of the FRAMES proposal shown in Figure 4.

Chaudhuri-Hocquenghem (BCH) coding combined with an intelligent burst-by-burst adaptive wideband multi-mode Quadrature Amplitude Modulation (QAM) modem, which can be configured either under network control or under transceiver control to operate as a 1, 2, 4 and 6 bits/symbol scheme, while maintaining a constant signalling rate. This allowed us to support an increased throughput expressed in terms of the average number of bits per symbol, when the instantaneous channel quality was high, leading ultimately to an increased video quality in a constant bandwidth.

The transmitted bitrate for all four modes of operation is shown in Table 3. The unprotected bitrate before approximately half-rate BCH coding is also shown in Table 3. The actual useful bitrate available for video is slightly less; than the unprotected bitrate due to the required strongly protected packet acknowledgement information and packetisation information. The effective video bitrate is also shown in Table 3.

4.4 Burst-by-Burst Adaptive Videophone Performance

The described burst-by-burst adaptive modem maximizes the system capacity available by using the most appropriate modulation mode for the current instantaneous channel conditions. We found that the pseudo-SNR at the output of the channel equaliser was an adequate channel quality measure in our burst-

Features	Multi-rate System			
Mode	BPSK	4QAM	16QAM	64QAM
Bits/Symbol	1	2	4	6
FEC	Near Half-rate BCH			
Transmission bitrate (kbit/s)	148.2	296.4	592.8	889.3
Unprotected bitrate (kbit/s)	75.8	151.7	303.4	456.1
Effective Video-rate (kbit/s)	67.0	141.7	292.1	446.4
Video fr. rate (Hz)	30			

Table 3: Operational-mode specific transceiver parameters

254 by-burst adaptive wide-band modem. Figure 5 demonstrates how the burst-by-burst adaptive modem
 255 changes its modulation modes every transmission burst, ie every 4.615 ms, based on the fluctuating
 256 pseudo-SNR. The right-hand-side vertical axis indicates the associated number of bits per symbol.
 257 By changing to more robust modulation schemes automatically, when the channel quality is reduced
 258 allows the packet loss ratio, or synonymously, the FER, to be reduced, which results in increased per-
 259 ceived video quality. In order to judge the benefits of burst-by-burst adaptive modulation we considered
 260 two scenarios. In the first scheme the adaptive modem always chose the perfectly estimated AQAM
 261 modulation mode, in order to provide a maximum upper bound performance. In the second scenario
 262 the modulation mode was based upon the perfectly estimated AQAM modulation mode for the previous
 263 burst, which corresponded to a delay of one Time Division Multiple Access (TDMA) frame duration
 264 of 4.615ms. This second scenario represents a practical burst-by-burst adaptive modem, where the one-
 265 frame channel quality estimation latency is due to superimposing the receiver's perceived channel quality
 266 on a reverse-direction packet, for informing the transmitter concerning the best mode to be used.
 267 The probability of the adaptive modem using each modulation mode for a particular average channel
 268 SNR is portrayed in Figure 6 in terms of the associated modem mode probability density functions
 269 (PDFs). It can be seen at high channel SNRs that the modem mainly uses the 64QAM modulation mode,
 270 while at low channel SNRs the BPSK mode is the most prevalent one.
 271 Figure 7 shows the transmission FER (or packet loss ratio) versus channel SNR for the 1, 2, 4 and 6
 272 bit/symbol fixed modulation schemes, as well as for the ideal and for the one-frame delayed realistic

273 scenarios using the burst-by-burst adaptive QAM (AQAM) modem. In the ideal - ie zero-delay - AQAM
274 scenario, where the modulation mode estimation is assumed to be available instantaneously, the trans-
275 mission FER is zero at high channel SNRs even though 64QAM is used predominantly, while at low
276 SNRs it exhibits a similar FER behaviour to fixed BPSK modulation, since this is the most often used
277 mode. More explicitly, at high SNRs the adaptive modem chooses the most suitable AQAM mode and
278 hence no packets are lost. However, at low SNRs the adaptive modem opts for using BPSK, even when
279 the channel is so hostile that the packets are lost. Hence the BPSK and ideal - ie zero-delay - AQAM
280 results are very similar at low channel SNRs. However, when the modulation mode estimation is delayed
281 by one TDMA frame - representing a realistic, practical AQAM modem - then the transmission FER is
282 no longer zero at high channel SNRs, since the delay results in a non-optimum modulation mode selec-
283 tion, which can result in the corresponding video packet being lost. At high channel SNRs the FER of
284 the realistic, one-frame delay AQAM modem is similar to that of the fixed 64QAM modem mode. By
285 contrast, at low channel SNRs its FER performance is similar to that of the fixed BPSK modem mode.
286 However, at medium channel SNRs the transmission FER is almost constant at about 3% for the realistic
287 AQAM modem. This is more clearly demonstrated on a logarithmic scale in Figure 8.

288 Explicitly, the ideal and realistic AQAM modems are compared to a fixed modulation based, statically
289 re-configured multi-mode system with switching at 5% transmission FER in Figure 8. The statically
290 reconfigured modem was invoked here as a benchmark, in order to indicate, how a system would
291 perform, which cannot act on the basis of the near-instantaneously varying channel quality. As it can
292 be inferred from Figure 8, such a statically reconfigured transceiver switches its mode of operation from
293 a lower-order modem mode, such as for example BPSK to a higher-order mode, such as 4QAM, when
294 the channel quality has improved sufficiently for the 4QAM mode's FER to become lower than 5 %
295 upon reconfiguring the transceiver in this 4QAM mode. Again - as seen in Figure 7 earlier on a non-
296 logarithmic scale - Figure 8 clearly shows that the ideal AQAM modem has a similar FER performance
297 to the fixed rate BPSK modem. Additionally, it indicates that the realistic AQAM modem has a similar
298 FER performance to the BPSK modem at low SNRs, yielding a near-constant 3% FER at medium SNRs
299 and a FER similar to that of the fixed 64QAM modem at high channel SNRs.

300 A comparison of the effective video bitrate versus channel SNR is shown in Figure 9 for the four fixed
301 modulation schemes, and the ideal and realistic AQAM modems. The effective video bitrate is the
302 average bitrate provided by all the successfully transmitted video packets. It should be noted that the
303 realistic AQAM modem has a slightly lower throughput, since sometimes the incorrect modulation mode
304 is chosen, which may result in packet loss. At very low channel SNRs the throughput bitrate converges
305 to that of the fixed BPSK mode, since the AQAM modem is almost always in the BPSK mode at these

306 SNRs, as demonstrated in Figure 6.

307 Having shown the effect of the burst-by-burst adaptive modem on the transmission FER and effective

308 bitrate, let us now demonstrate these effects on the decoded video quality, measured in terms of the Peak

309 Signal-to-Noise Ratio (PSNR). Figure 10 shows the decoded video quality in terms of PSNR versus

310 channel SNR for both the ideal and realistic adaptive modem, and for the four modes of the statically

311 configured multi-mode system. It can be seen that - as expected - the ideal adaptive modem, which

312 always selects the best of the modulation modes, has a better or similar video quality for the whole range

313 of channel SNR. In the statically configured multi-mode scheme the video quality degrades, when

314 the system switches from a higher-order to a lower-order modulation mode. The ideal adaptive modem

315 however smoothes out the sudden loss of video quality, as the channel degrades. The non-ideal adaptive

316 modem has a slightly lower video quality performance, than the ideal adaptive modem, especially at

317 medium SNRs, since it sometimes selects the incorrect modulation mode due to the estimation delay.

318 This can inflict video packet loss and/or a reduction of the effective video bitrate, which in turn reduces

319 the video quality.

320 The difference between the ideal burst-by-adaptive modem, using ideal channel estimation and the non-

321 ideal, realistic burst-by-burst adaptive modem, employing a non-ideal delayed channel estimation can be

322 seen more clearly in Figure 11 for a range of video sequences. Observe that at high and low channel

323 SNRs the video quality performance is similar for the ideal and non-ideal adaptive modems. This is

324 because the channel estimation delay has little effect, since at low or high channel SNRs the adaptive

325 modem is in either BPSK or 64QAM mode for the majority of the time. More explicitly, the channel

326 quality of a transmission frame is almost always the same as that of the next, and hence the delay has

327 little effect at low and high SNRs.

328 The video quality versus channel quality trade-offs can be more explicitly observed in Figure 12. This

329 figure portrays the decoded video quality in PSNR versus the packet loss ratio or transmission FER.

330 The ideal and practical adaptive modem performance is plotted against that of the four fixed modulation

331 schemes in the figure. It can be seen that the adaptive modems' video quality degrades from that

332 achieved by the error free 64QAM modem towards the BPSK modem performance as the packet loss

333 ratio increases. The practical adaptive modems' near constant FER performance of 3% at medium SNRs

334 can be clearly seen in the figure, which is associated with the reduced PSNRs of the various modem

335 modes, while having only minor channel error-induced impairments.

	BPSK	4QAM	16QAM	64QAM
Standard	<10dB	≥ 10 dB	≥ 18 dB	≥ 24 dB
Conservative	<13dB	≥ 13 dB	≥ 20 dB	≥ 26 dB
Aggressive	<9dB	≥ 9 dB	≥ 17 dB	≥ 23 dB

Table 4: SINR estimate at output of the equaliser required for each modulation mode in Burst-by-Burst Adaptive modem, ie. switching thresholds

4.5 Switching Thresholds

The burst-by-burst adaptive modem changes its modulation modes based on the fluctuating channel conditions expressed in terms of the SNR at the equaliser's output. The set of switching thresholds used in all the previous graphs is the "Standard" set shown in Table 4, which was determined on the basis of the required channel SINR for maintaining the specific target video FER.

In order to investigate the effect of different sets of switching thresholds, we defined two new sets of thresholds, a more conservative set, and a more aggressive set, employing less robust, but more bandwidth-efficient modem modes at lower SNRs. The more conservative switching thresholds reduced the transmission FER at the expense of a lower effective video bitrate. The more aggressive set of thresholds increased the effective video bitrate at the expense of a higher transmission FER.

The transmission FER performance of the realistic burst-by-burst adaptive modem, which has a one TDMA frame delay between channel quality estimation and mode switching is shown in Figure 13 for the three sets of switching thresholds of Table 4. It can be seen that the more conservative switching thresholds reduce the transmission FER from about 3% to about 1% for medium channel SNRs. The more aggressive switching thresholds increase the transmission FER from about 3% to 4-5%. However, since FERs below 5% are not objectionable in video quality terms, this FER increase is an acceptable compromise for a higher effective video bitrate. The effective video bitrate for the realistic adaptive modem with the three sets of switching thresholds is shown in Figure 14. The more conservative set of switching thresholds reduces the effective video bitrate but also reduces the transmission FER. The aggressive switching thresholds, increase the effective video bitrate, but also increase the transmission FER. Therefore the optimal switching thresholds should be set such that the transmission FER is deemed acceptable is the range of channel SNRs considered.

5 Summary

The above-described burst-by-burst adaptive multimedia transceiver concept exhibits substantial advantages in comparison to conventional fixed-mode or statically reconfigurable transceivers, which was substantiated in the context of a specific embodiment of the advocated system concept, namely with the aid of a burst-by-burst adaptive video transceiver.

Specifically, the main advantage of the described burst-by-burst adaptive transceiver technique is that irrespective of the prevailing channel conditions, the transceiver achieves always the best possible source-signal representation quality - such as video, speech or audio quality - by automatically adjusting the achievable bitrate and the associated multimedia source-signal representation quality in order to match the channel quality experienced. This is achieved on a near-instantaneous or burst-by-burst adaptive basis under given propagation conditions in order to cater for the effects of path-loss, fast-fading, slow-fading, dispersion, co-channel interference, etc. Furthermore, when the mobile is roaming in a hostile out-doors - or even hilly terrain - propagation environment, typically low-order, low-rate modem modes are invoked, while in benign indoor environments predominantly the high-rate, high source-signal representation quality modes are employed.

The described system embodiment has the following features:

1. A reliable instantaneous channel quality metric is employed, in order to appropriately configure the AQAM modem for maintaining the required target BER and the associated source signal representation quality. The range of potential channel quality metrics entails the pseudo-SNR, SINR, BER and its LLR-based channel estimates.
2. The perceived channel quality determines the number of bits that can be transmitted in a given transmitted packet or burst, which in turn predetermines the number of bits to be generated by the associated multimedia source codec, such as for example the associated video, audio, speech or handwriting codec. Hence the multimedia source codec has to be capable of adjusting the number of bits generated under the instruction of the burst-by-burst adaptive transceiver.
3. The transmitter mode requested by the receiver, in order to achieve the target performance has to be signalled by the receiver to the remote transmitter. Another scenario is, where the uplink and downlink channel quality is sufficiently similar for allowing the receiver to judge, what transmission mode the associated transmitter should use, in order for its transmitted signal to maintain the required transmission integrity. Lastly, the mode of operation used by the transmitter can also be detected using blind detection techniques, for example in conjunction with the associated channel

389 decoder.

390 In the studied example of the system embodiment we have characterised a wideband burst-by-burst adap-
391 tive multimedia transceiver which employed the pseudo-SNR at the output of the channel equaliser as
392 the quality measure for controlling the AQAM modem modes. Whilst in reference [16] the through-
393 put upper-bound of such an AQAM modem was analysed, in this document a practical multimedia
394 transceiver concept was described and the achievable performance gains due to employing the described
395 wideband burst-by-burst adaptive modem were quantified. An adaptive packetiser was used in conjunc-
396 tion with the adaptive modem, which continually adjusted the video codec's target bitrate, in order to
397 exploit the instantaneous bitrate provided by the adaptive modem.

398 In the example the delay between the channel estimation and modulation mode switching was shown
399 to have a considerable effect on the performance achieved by the adaptive modem. This performance
400 penalty can be mitigated by reducing the modem mode switching latency, for example by employing
401 adjacent slots for the uplink and downlink of a TDD system. However, at lower vehicular speeds
402 the effects of AQAM mode switching latency are less crucial and the practical adaptive modem can
403 achieve a performance that is close to that of the ideal adaptive modem exhibiting no switching latency,
404 that we used as an upper bound benchmark. We have also demonstrated, how the transmission FER
405 performance is affected by changing the switching thresholds. Therefore the system can be tuned to the
406 required FER performance using appropriate switching thresholds.

407 It will be appreciated that although a particular embodiment of the invention has been described, many
408 modifications, additions and/or substitutions may be made within the spirit and scope of the present
409 invention

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CLAIMS

1. A receiver unit comprising:

a burst-by-burst adaptive equalizer having an input for receiving data bursts from a
5 communication channel, each burst containing a number of bits per symbol;

a computational unit for computing a received signal quality metric related to a bit
error rate experienced during transmission over the communication channel;

10 an output for relaying the signal quality metric, conveying signal quality as perceived
by the receiver unit, for use in determining a configuration for subsequent transmission
bursts.
2. A receiver unit according to claim 1, wherein the received signal quality metric
15 is evaluated from an interference parameter.
3. A receiver unit according to claim 2, wherein the signal quality metric is
evaluated using channel impulse response estimates derived from a training sequence
embedded in each transmission burst.
20
4. A receiver unit according to claim 2, wherein the interference parameter is a
measure of co-channel interference.
5. A receiver unit according to claim 2, wherein the interference parameter is a
25 measure of inter-symbol interference.

6. A receiver unit according to claim 1, wherein the signal quality metric is evaluated according to the formula:

$$\begin{aligned}
 \gamma_{dfe} &= \frac{\text{Wanted Signal Power}}{\text{Residual ISI Power} + \text{Effective Noise Power}} \\
 &= \frac{E \left[|S_k \sum_{m=0}^{N_f-1} C_m h_m|^2 \right]}{\sum_{q=-(N_f-1)}^{-1} E \left[|f_q S_{k-q}|^2 \right] + N_0 \sum_{m=0}^{N_f-1} |C_m|^2}
 \end{aligned}$$

10 where γ_{dfe} is pseudo-SNR at the output of the equalizer, C_m and H_m denote feed-forward coefficients and the channel impulse response respectively, S_k and N_0 are transmitted signal and noise spectral density respectively, and N_f is the number of feed-forward coefficients.

15 7. A receiver unit according to claim 1, wherein the received signal quality metric is evaluated from the bit error rate.

8. A receiver unit according to claim 7, wherein the bit error rate is estimated by an algebraic channel decoder.

20

9. A receiver unit according to claim 7, further comprising a channel decoder arranged to receive the data bursts from the adaptive equalizer, and wherein the bit error rate is estimated from a calculation of a logarithmic likelihood ratio, thereby to provide a reliable estimator of all possible channel impairments.

25

10. A receiver unit according to claim 9, wherein the logarithmic likelihood ratio is calculated at the input of the channel decoder.

11. A receiver unit according to claim 9, wherein the logarithmic likelihood ratio is calculated at the output of the channel decoder.

30

12. A receiver unit according to any one of the preceding claims, wherein the configuration defines the number of bits per symbol in each transmission burst, which is varied according to the signal quality metric computed from a previous transmission burst, as supplied by the output.

5

13. A receiver unit according to any one of the preceding claims, wherein the output is arranged to relay the signal quality metric, representative of signal quality as perceived by the receiver unit, through the communication channel for configuration of the remote transmitter for subsequent transmission bursts, thereby to provide closed-loop feedback.

10

14. A receiver unit according to any one of claims 1 to 12, wherein the output is arranged to relay the signal quality metric, representative of signal quality as perceived by the receiver unit, to a transmitter unit local to the receiver unit for configuration of the local transmitter unit for subsequent transmission bursts to a remote receiver unit associated with the remote transmitter unit, thereby to provide open-loop feedback.

15

15. A receiver unit according to any one of claims 1 to 12, wherein the signal quality metric is internally used in a blind detection scheme to reconfigure the receiver unit for decoding subsequent transmission bursts.

20

16. A system comprising a receiver unit according to any one of claims 1 to 12 in combination with a transmitter unit, wherein the transmitter unit has an input connected to the output of the receiver unit for receiving the signal quality metric, the transmitter unit having a configuration that is responsive to the signal quality metric.

25

17. A system according to claim 16, wherein the transmitter unit comprises an interactive multimedia encoder having a configuration that is responsive to the signal quality metric.

30

18. A system according to claim 16 or 17, wherein the transmitter unit comprises a modem having a configuration that is responsive to the signal quality metric.

19. A system according to claim 16, 17 or 18, wherein the transmitter unit comprises a channel encoder having a configuration that is responsive to the signal quality metric.

5

20. A system according to any one of claims 16 to 19, wherein the transmitter unit and receiver unit form a transceiver unit.

21. A system according to any one of claims 16 to 19, wherein the transmitter unit and receiver unit are remote from each other and form respective parts of separate transceiver units.

15

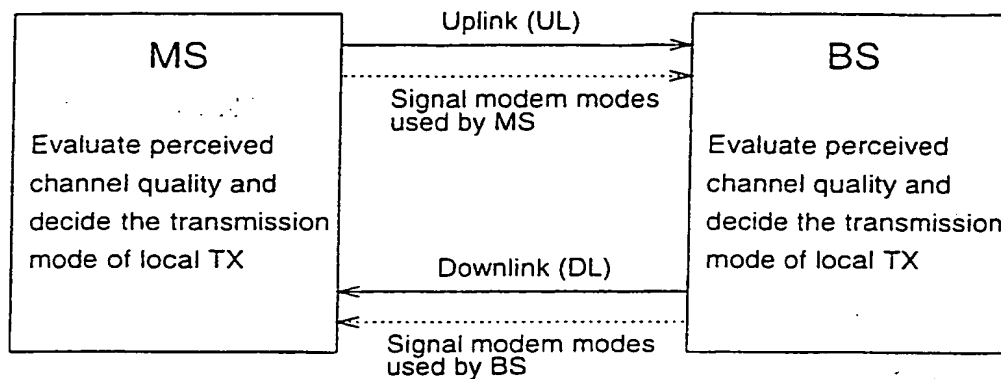


Figure 1(a)

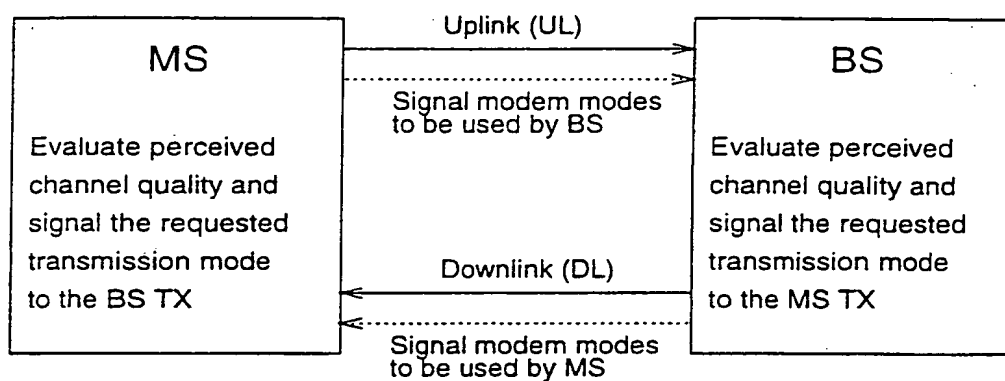


Figure 1(b)

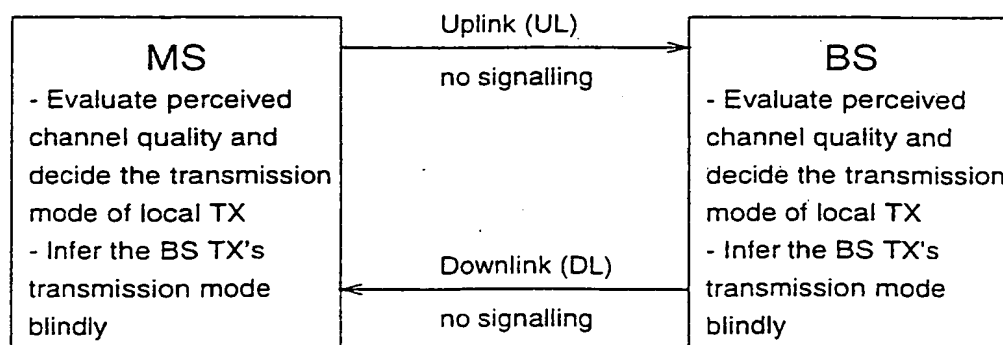


Figure 1(c)

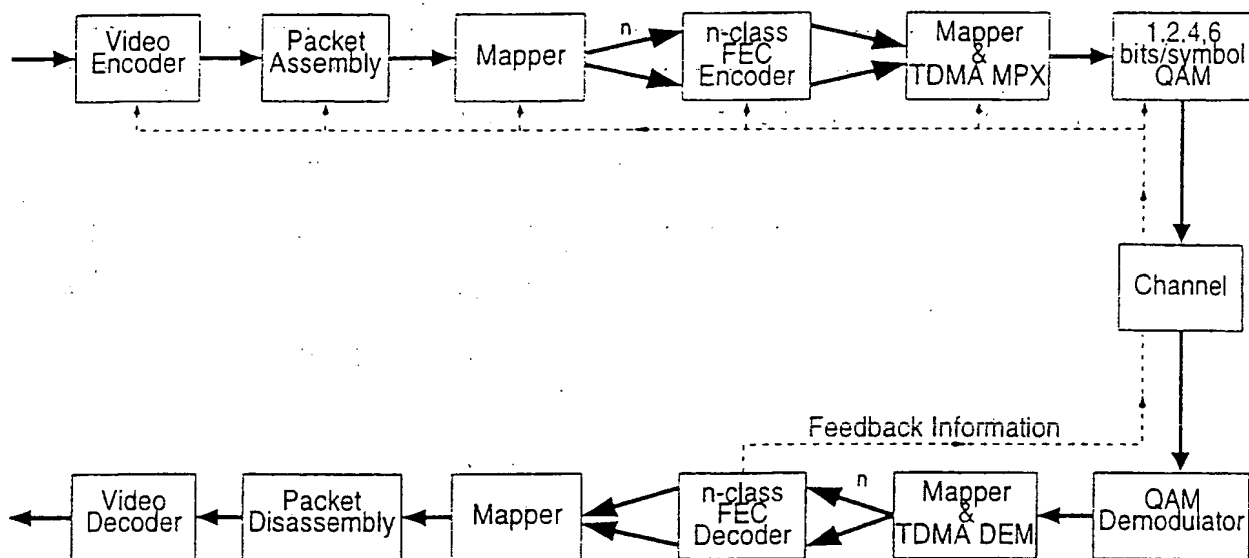


Figure 2

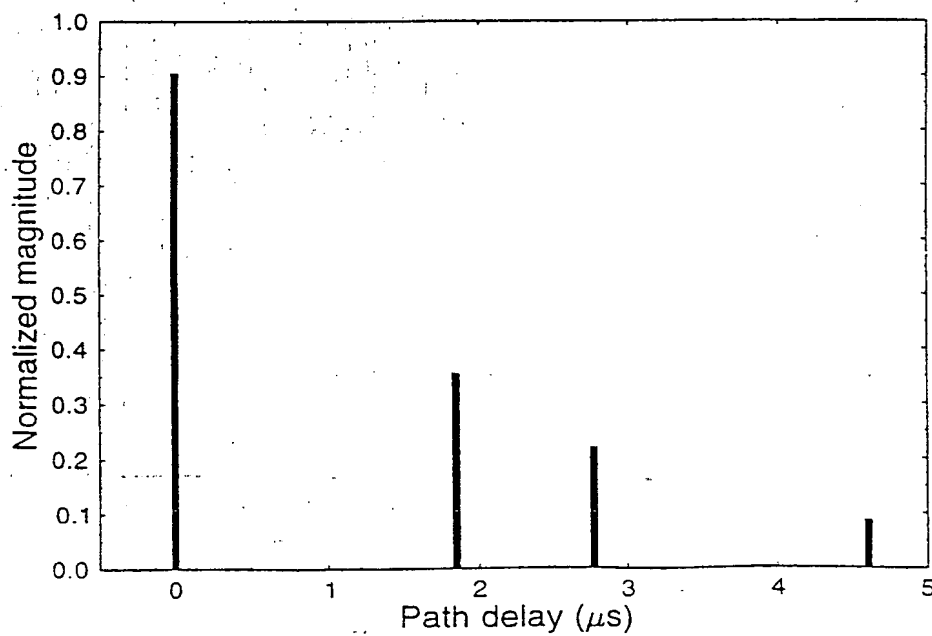


Figure 3

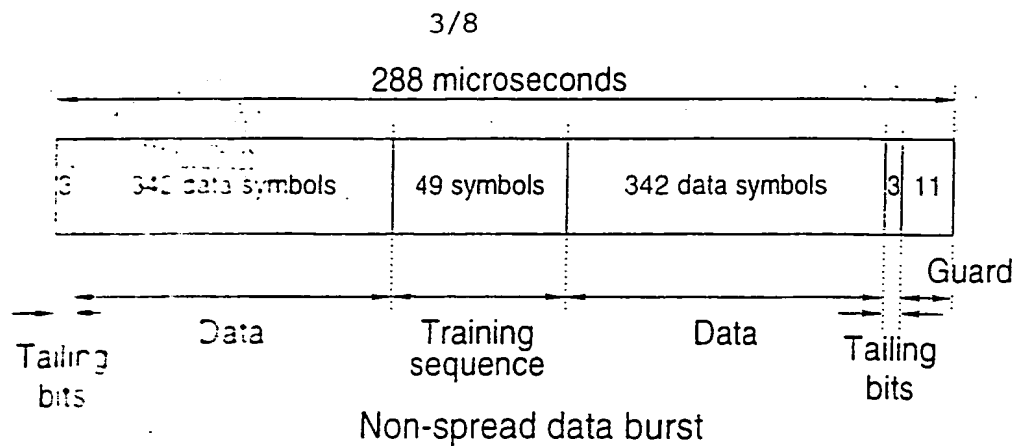


Figure 4

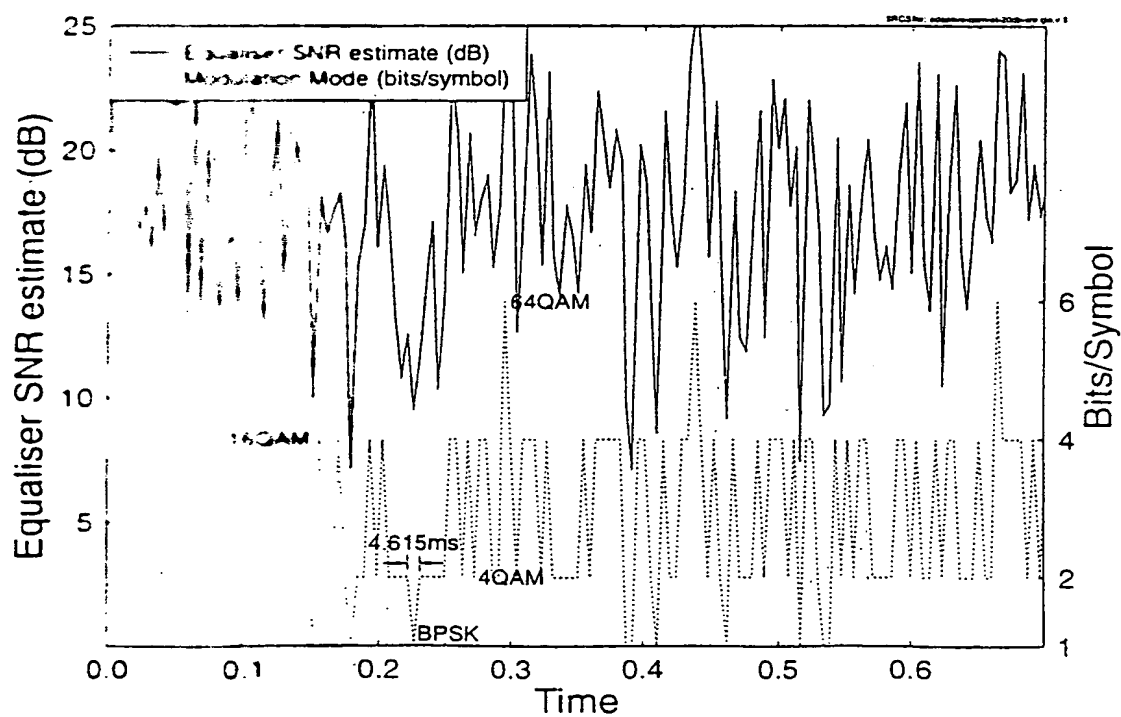


Figure 5

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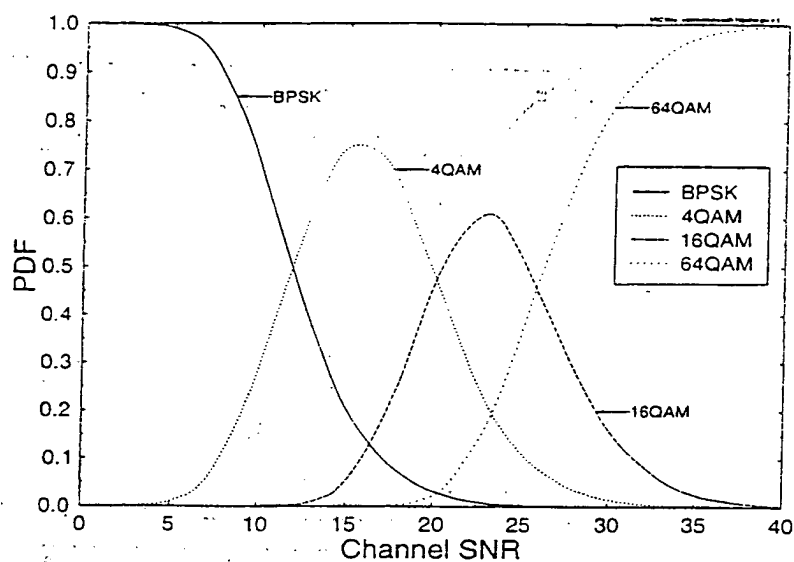


Figure 6

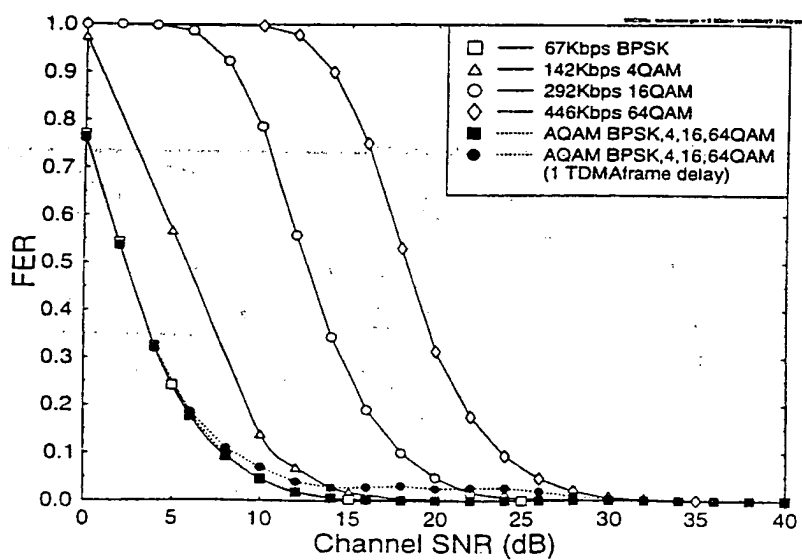


Figure 7

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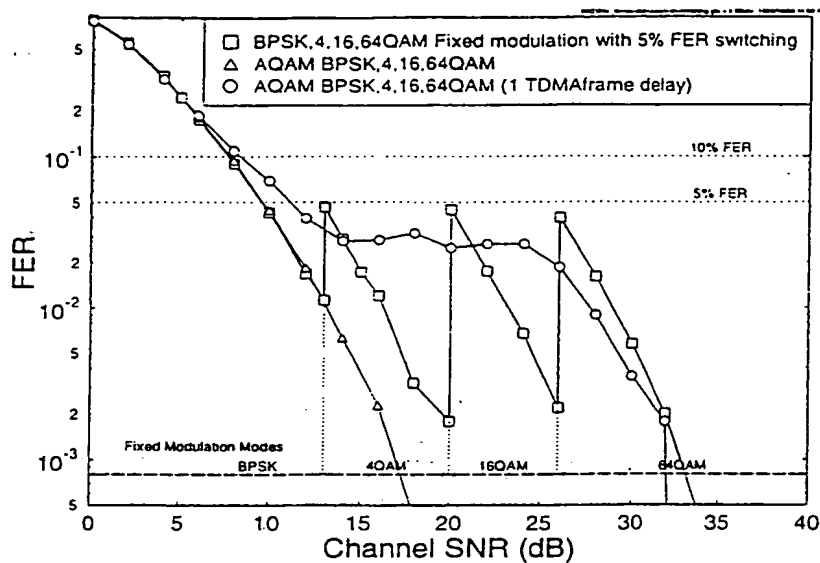


Figure 8

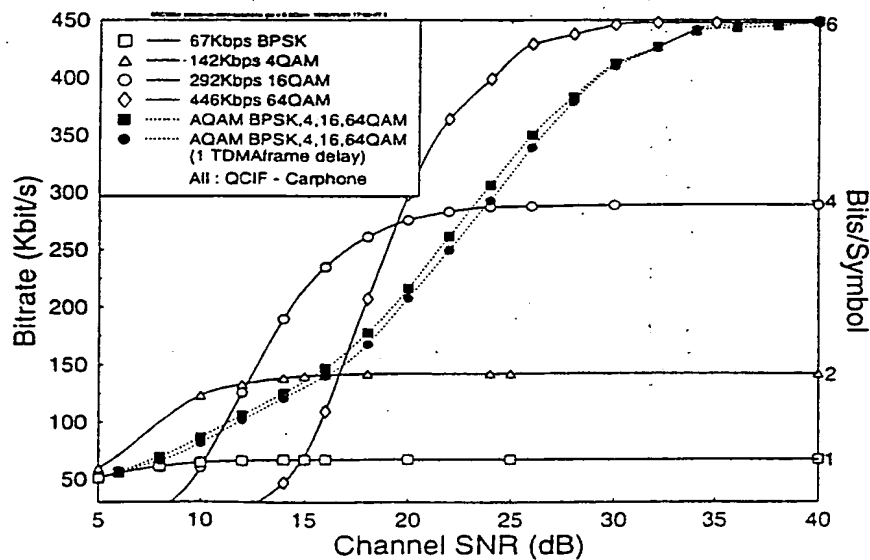


Figure 9

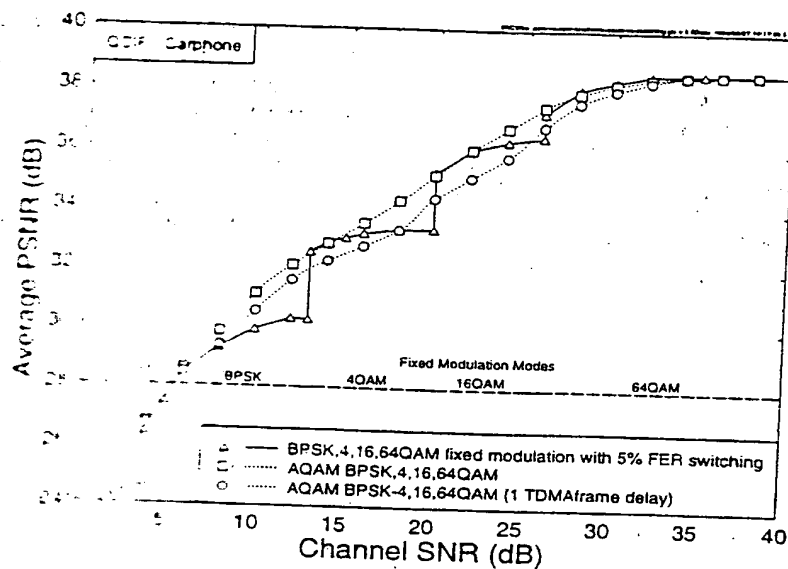


Figure 10

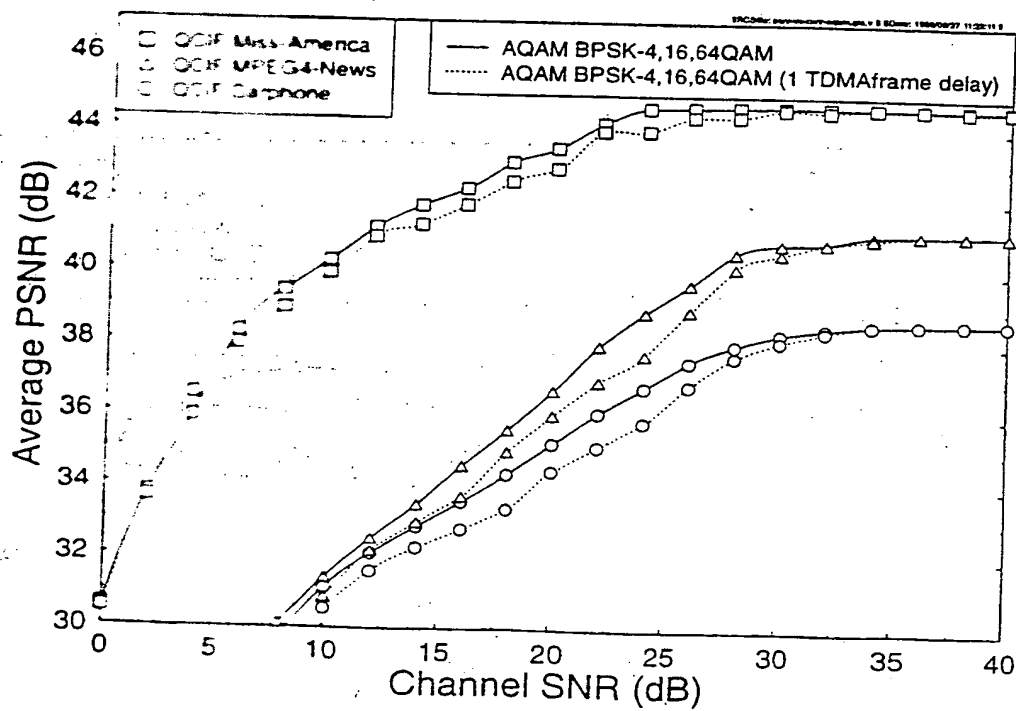


Figure 11

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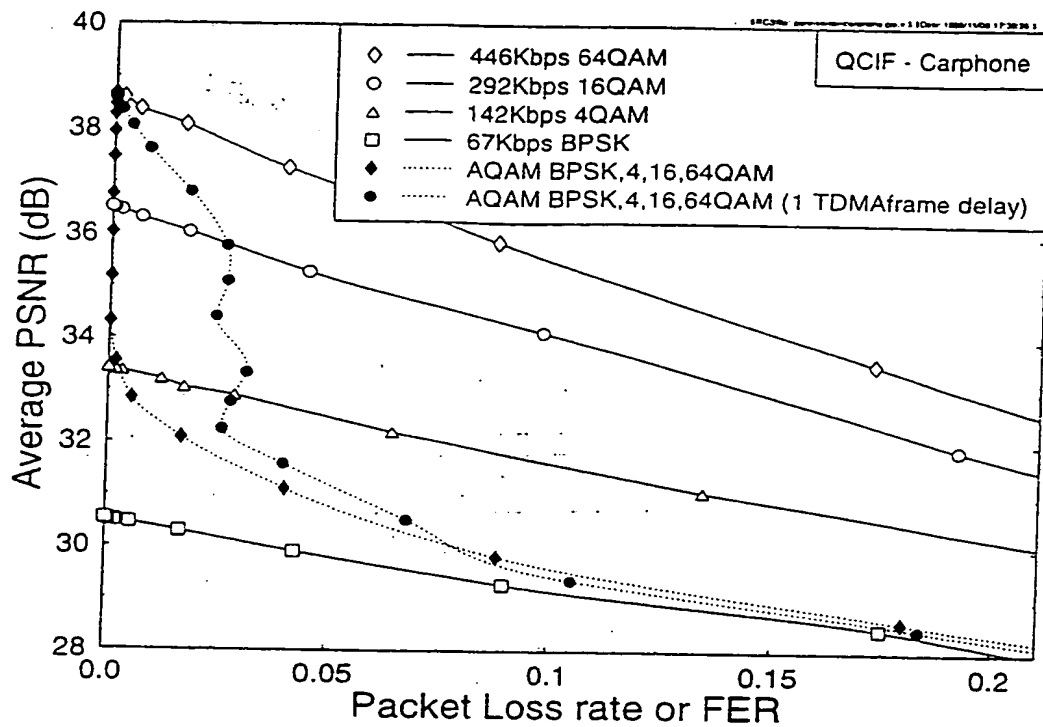


Figure 12

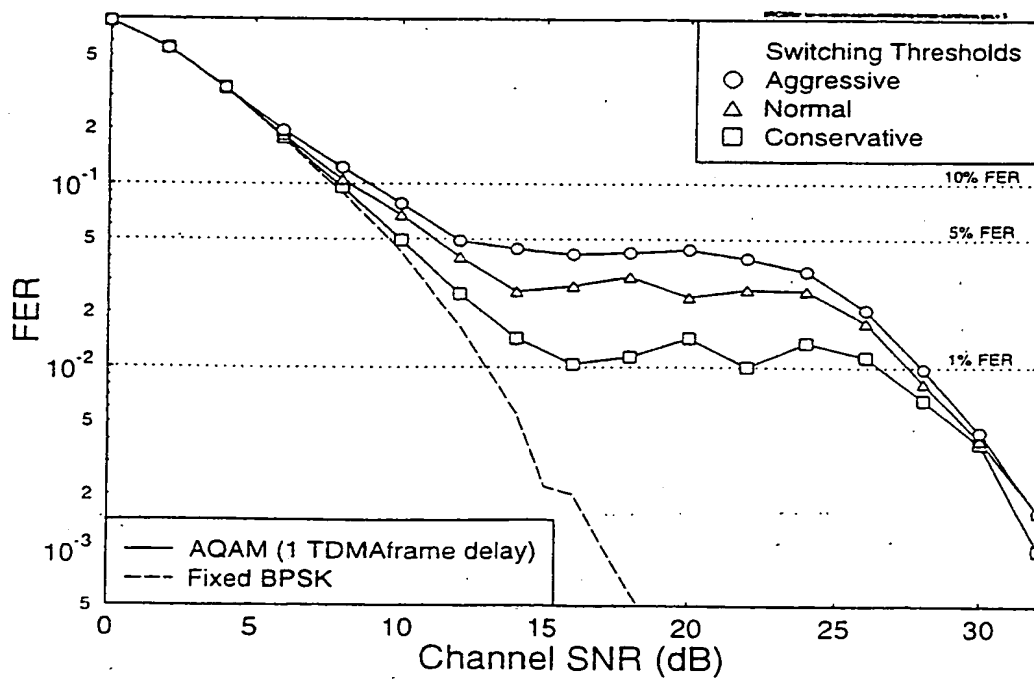


Figure 13

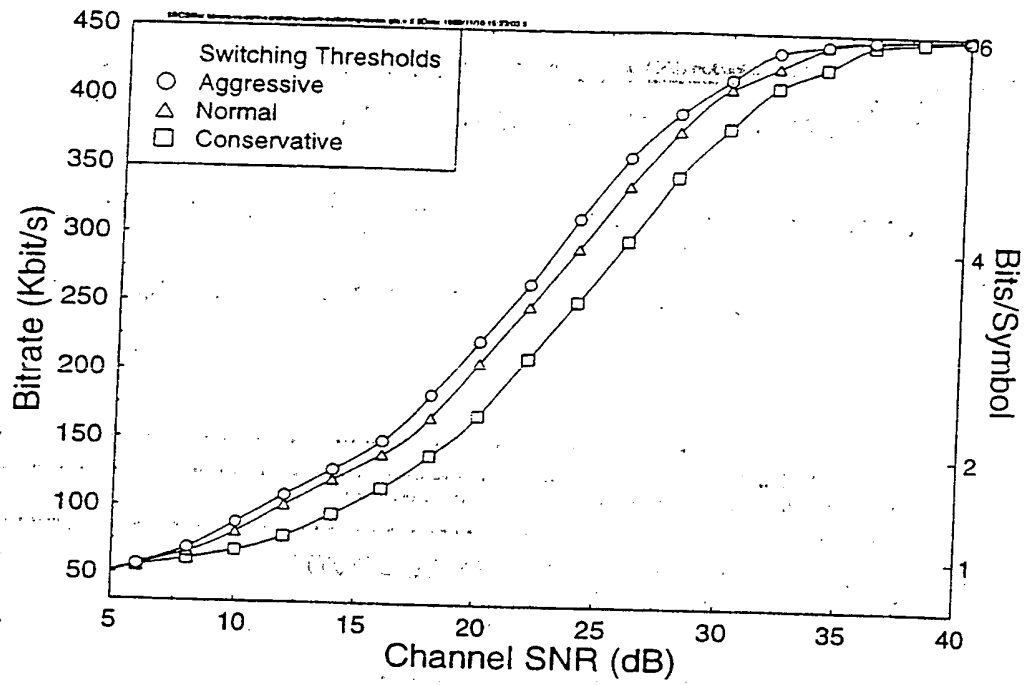


Figure 14

INTERNATIONAL SEARCH REPORT

International Application No
PCT/GB 00/00017

A. CLASSIFICATION OF SUBJECT MATTER

IPC 7 H04B1/10

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

IPC 7 H04B

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	<p>WO 98 51111 A (KONINKL PHILIPS ELECTRONICS NV ;PHILIPS AB (SE)) 12 November 1998 (1998-11-12)</p> <p>page 14, line 30 -page 16, line 31; figures 1,8 claims 6,14 page 5, line 30 -page 6, line 14 abstract</p> <p style="text-align: center;">— -/-</p>	<p>1,7, 12-14, 16-18, 20,21</p>

☒ Further documents are listed in the continuation of box C.

☒ Patent family members are listed in annex.

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- "&" document member of the same patent family

Date of the actual completion of the international search

17 February 2000

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Kolbe, W

INTERNATIONAL SEARCH REPORT

International Application No
PCT/GB 00/00017

C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT		
Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y	<p>OTSUKI S ET AL: "PERFORMANCE OF MODULATION-LEVEL-CONTROLLED ADAPTIVE MODULATION SYSTEMS" ELECTRONICS & COMMUNICATIONS IN JAPAN, PART I - COMMUNICATIONS,US,SCRIPTA TECHNICA. NEW YORK, vol. 79, no. 7, 1 July 1996 (1996-07-01), pages 81-93, XP000696376 ISSN: 8756-6621 page 82, right-hand column, line 22 -page 84, left-hand column, line 21; figures 1,2</p>	1,16
Y	<p>SEIICHI SAMPEI ET AL: "ADAPTIVE MODULATION/TDMA WITH A BDDFE FOR 2 MBIT/S MULTI-MEDIA WIRELESS COMMUNICATION SYTEMS" PROCEEDINGS OF THE VEHICULAR TECHNOLOGY CONFERENCE,US,NEW YORK, IEEE, vol. CONF. 45, 1995, pages 311-315, XP000550185 ISBN: 0-7803-2743-8 page 311, right-hand column, line 6 -page 312, right-hand column, line 4; figure 1</p>	1,16
A	<p>EP 0 878 924 A (YRP MOBILE TELECOMMUNICATIONS ;HITACHI LTD (JP); SHARP KK (JP); CO) 18 November 1998 (1998-11-18) page 7, line 15 - line 30 abstract; figures 1,2</p>	1

INTERNATIONAL SEARCH REPORT

information on patent family members

Intern: 1st Application No
PCT/GB 00/00017

Patent document cited in search report		Publication date		Patent family member(s)	Publication date
WO 9851111	A	12-11-1998	EP	0935903 A	18-08-1999
EP 0878924	A	18-11-1998	JP	10313274 A	24-11-1998

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